Power amplifiers and the use of pulse modulation

Amplifier types

• Voltage amplification

• Current amplification (buffer, driver)

• Power amplification
Amplifier characteristics

- Single sided power supply
- Double sided power supply

The active element (transistor) must be biased to work in its active region

Amplifier classes

- Class A
- Class B
- Class AB
- Class C
- Class D
- Class G
- Class H

"Digital" This class uses pulse modulation

Analog
Class A

Base bias gives quiescent point

Quiescent point

Remove DC from load (HP filter)

Class A cont.
Class B

In rest
\[ U = 0 \]
\[ I = 0 \]

No need to remove DC for a double sided power supply

Crossover distortion
Class AB

Bias diodes gives quiescent points
This circuit is highly simplified

Class C

\[ f_c = \frac{1}{2 \cdot \pi \cdot \sqrt{L \cdot C}} \]

Tuned circuit
Reasonably linear in narrow band
Carrier wave of radio signal
Class G

Adapt the power supply voltage by making it switchable in level

In some cases also use separate power transistors for the separate voltage levels

Class H

Dynamically adapt the power supply voltage by modulating it in accordance with the input voltage

Used in power amps manufactured by Labgruppen in Kungsbacka
Switching amplifier, Class D

Fully on (saturated) means
to low voltage
high current

Fully off means
high voltage
no current

In both cases power
dissipation $U \cdot I$ is small

Pulse width modulation (PWM)

Pulse train with fixed frequency

Modulation of the duty cycle

Duty cycle $= \frac{\text{pulse time}}{\text{period time}} \cdot 100\%$

High input signal $\rightarrow$ high duty cycle
Low input signal $\rightarrow$ low duty cycle
Pulse width modulation (PWM)

Duty cycle

- 0%
- 50%
- 75%
- 100%

Pulse width modulation (PWM)

PWM generation
Demonstration

Left aligned channels

Periods start with level changes at the same time
Center aligned channels

Periods don’t start with level changes at the same time unless the duty cycles are the same

Observe the doubled period time

Pulse density modulation (PDM)

Similar to PWM

Fixed period time

The whole period is either high or low
Pulse density modulation (PDM)

Definitions:
- Pulse time
- Amplitude
- Period time
- Pulse
- Pulse gap
Frequency analysis of PWM signal

$$f(t) = A \cdot \frac{T}{T} + 4 \cdot A \cdot \frac{T}{T} \sum_{k=1}^{\infty} \sin \left( \frac{k \cdot \omega_0 \cdot T}{2} \right) \cos(k \cdot \omega_0 \cdot t)$$

Amplitude of frequency component

DC component

Frequency components

Fundamental frequency $f_0$

Frequency $k \cdot f_0$

Lowpass LC filter

$$\omega_0 = 2 \cdot \pi \cdot f_0 = \frac{2 \cdot \pi}{T}$$
Lowpass LC filter cont.

\[ H(\omega) = \frac{R \frac{1}{j \omega C}}{j \omega L + R \frac{1}{j \omega C}} = \frac{1}{1 - \omega^2 L C + j \frac{\omega L}{R}} \]

\[ \omega = 2 \pi f \]

Second order filter equations

**Lowpass filter**

\[ H_{lp}(\omega) = H_{io} \cdot \frac{1}{1 - \left(\frac{\omega}{\omega_0}\right)^2 + j \frac{\omega}{Q \cdot \omega_0}} \]

**Highpass filter**

\[ H_{hp}(\omega) = H_{io} \cdot \left(\frac{\omega}{\omega_0}\right)^2 \]

**Bandpass filter**

\[ H_{bp}(\omega) = H_{io} \cdot \left(\frac{\omega}{\omega_0}\right)^2 + j \frac{\omega}{Q \cdot \omega_0} \]

**Bandreject filter**

\[ H_{br}(\omega) = H_{io} \cdot \frac{\omega}{\omega_0} \]
Lowpass LC filter cont.

Second order filter equations

### Lowpass filter

\[
H_{lp}(f) = H_{lp} \cdot \frac{1}{1 - \left( \frac{f}{f_0} \right)^2 + j \cdot \frac{f}{Q \cdot f_0}}
\]

### Highpass filter

\[
H_{hp}(f) = H_{hp} \cdot \frac{-\left( \frac{f}{f_0} \right)^2}{1 - \left( \frac{f}{f_0} \right)^2 + j \cdot \frac{f}{Q \cdot f_0}}
\]

### Bandpass filter

\[
H_{bp}(f) = H_{bp} \cdot \frac{j \cdot \frac{f}{Q \cdot f_0}}{1 - \left( \frac{f}{f_0} \right)^2 + j \cdot \frac{f}{Q \cdot f_0}}
\]

### Bandreject filter

\[
H_{br}(f) = H_{br} \cdot \frac{1 - \left( \frac{f}{f_0} \right)^2}{1 - \left( \frac{f}{f_0} \right)^2 + j \cdot \frac{f}{Q \cdot f_0}}
\]

### Lowpass LC filter cont.

\[
H(\omega) = \frac{1}{1 - \omega^2 \cdot L \cdot C + j \cdot \frac{\omega}{R}} = \frac{1}{1 - \left( \frac{\omega}{\omega_0} \right)^2 + j \cdot \frac{\omega}{Q \cdot \omega_0}}
\]

\[
\omega = 2 \cdot \pi \cdot f
\]

Identify constants

- Cutoff frequency: \( f_0 = \frac{1}{2 \cdot \pi \cdot \sqrt{L \cdot C}} \)
- Q value: \( Q = \frac{R \cdot \sqrt{C}}{\sqrt{L}} \)
Lowpass LC filter cont.

Demonstration
Low pass LC filter cont.

Active second order filter

\[ U_{\text{in}} - U_{\text{1}} + \frac{U_{\text{out}} - U_{\text{1}}}{1 + \frac{1}{j\omega C_2}} + \frac{U_{\text{2}} - U_{\text{1}}}{R_2} = 0 \]

\[ U_2 = \frac{1}{j\omega R_2 + \frac{1}{j\omega C_1}} \cdot U_1 = \frac{1}{1 + \frac{1}{j\omega R_2 C_1}} \cdot U_1 \]

\[ U_{\text{out}} = A \cdot U_2 \]
Active second order filter cont.

\[ \frac{U_{\text{out}}}{U_{\text{in}}} = \frac{A}{1 - \omega^2 \cdot R_1 \cdot R_2 \cdot C_1 \cdot C_2 + j \cdot \omega \cdot [R_1 (R_1 + R_2) \cdot C_1 + (1 - A) \cdot R_1 \cdot C_2]} \]

If \( A = 1 \)

\[ \frac{U_{\text{out}}}{U_{\text{in}}} = \frac{1}{1 - \omega^2 \cdot R_1 \cdot R_2 \cdot C_1 \cdot C_2 + j \cdot \omega \cdot (R_1 + R_2) \cdot C_1} = \frac{1}{1 - \left( \frac{\omega}{\omega_0} \right)^2 + j \frac{\omega}{Q \omega_0}} \]
Active second order filter cont.

\[ f_0 = \frac{\omega_0}{2\pi} = \frac{1}{2\pi \sqrt{R_1 R_2 C_1 C_2}} \]

\[ Q \cdot \omega_0 = \frac{1}{(R_1 + R_2) C_1} \Rightarrow Q = \frac{1}{R_1 + R_2} \sqrt{\frac{R_1 R_2 C_2}{C_1}} \]

Active second order filter cont.

If \( A = 1 \) and \( R_1 = R_2 = R \)

\[ \frac{U_{\text{out}}}{U_{\text{in}}} = \frac{1}{1 - \omega^2 R^2 C_1 C_2 + j \omega \cdot 2 \cdot R \cdot C_i} = \frac{1}{1 - \left(\frac{\omega}{\omega_0}\right)^2 + j \frac{\omega}{Q \cdot \omega_0}} \]
Active second order filter cont.

\[ f_0 = \frac{\omega_0}{2\pi} = \frac{1}{2\pi R \sqrt{C_1 C_2}} \]

\[ Q \cdot \omega_0 = \frac{1}{2 R C_1} \Rightarrow Q = \frac{1}{2} \sqrt{\frac{C_2}{C_1}} \]

Active second order filter cont.

This won’t work in our case!

Why?

The filter should be placed after the power amplifier and the operational amplifier can’t handle this power

We have to use a passive filter!
PWM-frequency vs. damping

Demonstration
PWM-frequency vs. resolution

n bits resolution → \( N = 2^n \) possible values (0 to N-1)

→ N possible duty cycles of the PWM-signal

Example 3 bits resolution → \( 2^3 = 8 \) possible values (0-7)

\[ PWM - frequency = \frac{\text{system clock frequency}}{7} \]

PWM-frequency vs. resolution cont.

We get the rule

\[ PWM - frequency = \frac{\text{system clock frequency}}{N - 1} = \frac{\text{system clock frequency}}{2^n - 1} \]

The higher the resolution the lower the PWM-frequency
PWM-frequency vs. damping and resolution

We could increase the resolution by increasing N.

In the same time the PWM period \((N-1)\cdot T\) gets longer.

That is the PWM-frequency falls.

And the PWM-frequency moves closer to the signal frequency.

It gets harder to filter out the pulse.
PWM-frequency vs. damping and resolution

We have to balance the PWM-frequency and the accuracy to fit our application

Filter problems

We need to attenuate the pulse out of our signal
High attenuation calls for higher order filters
It is hard to make passive filters of higher order
Tolerances makes the accuracy low
The filters tend to get bulky
mostly because of the inductors
Sigma-delta modulator

The pulse could be described as a noise floor on our signal

Remember digital-to-analog converters
Sigma-delta modulation cont.

Using sigma-delta modulation we can shape the noise and move it to higher frequencies where it gets more attenuated even using lower order filters.

![Diagram showing signal and noise power vs frequency](image)

3f_s/2

Typical PWM application

Speed control of electrical DC motors

The motor is a sluggish device that in itself acts as a lowpass filter

We don’t need no separate lowpass filter
Complete class D amplifier

Two types

• Amplifier without feedback

• Amplifier with feedback

Class D amplifier without feedback

$$f_s = 2f_{\text{signal max}}$$

Too low for PWM

Sampling frequency 44,1 kHz and 16 bits would give a system clock for PWM of $$2^{16} \cdot 44,1 \text{ kHz} = 2,89 \text{ GHz}$$ Too high

We lower the number of bits to 3-4

No control of the result without feedback
Class D amplifier with feedback

Noise damping and frequency correction in the passband

The signal we feed back is analog therefore ADC

The output signal has ripple therefore low-pass filter

Class D amplifier pros and cons

+ High efficiency
+ Easy integration to digital electronics
- The switching gives noise and distortion
- Noise on the power lines directly influence the output signal